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SYSTEM AND PROCESS FOR ROUTING TELEPHONY COMMUNICATIONS FROM  
A CONVENTIONAL TELEPHONE SET THROUGH A DATA NETWORK

DESCRIPTION

5 TECHNICAL FIELD:

The present invention relates to telecommunications systems and in particular to the routing of telephone communications. The invention is especially applicable to a system and a process for routing telephony communications from a conventional ("PSTN") telephone set through a public data network, such as the Internet, to terminate, optionally, at another  
10 conventional "PSTN" telephone; and to a subscriber interface unit for use therewith.

The invention is applicable whether access to the network is narrowband or broadband.

BACKGROUND ART

15 Known systems for routing conventional telephony communications through public data networks generally fall into one of three classes. The first class of conventional routing system, usually used by large companies with private communications links, uses "re-diallers". Re-diallers re-route specific calls through a separate transmission path, e.g. a data network, using a real-time transport mechanism. This system requires the re-dialler to have  
20 access to, and knowledge of, a private wide area data network, which is a significant disadvantage.

The second class of conventional routing system provides real-time voice communications from and to computer systems which have analog to digital hardware and appropriate software; the user using a headset. Recent modifications have allowed the use  
25 of a standard telephone set attached to the computer. These systems require the originator to establish a session with the data network, thus changing the user interface, and to have knowledge of the recipient's data address. Both users require a computer with special equipment and each user must have special knowledge, limiting the usefulness of this class of product to the general public. These are significant disadvantages.

30 The third class of conventional routing system provides real-time voice communications from and to gateways located between a private branch exchange (PBX) and

the public data network. This system requires a costly switching device between the PBX and the data network, thereby limiting its use to only enterprises of a significant size. This precludes use in a small business or home environment, which is a significant disadvantage.

## 5 DISCLOSURE OF INVENTION:

An object of the present invention is at least to ameliorate the disadvantages of such known systems, or at least provide an alternative.

According to a first aspect of the present invention, there is provided a method of routing telephone calls from a subscriber telephone (10) in a telecommunications system, comprising the Public Switched Telephone Network and a public data network, for example the Internet, to a called party, the system including a local central office (14), a data network entrance node (26), a data network exit point (30), a call server (38) provisioned with data and software for routing of data calls between the data network entrance node and the data network exit point, and a subscriber interface unit (12; 12A) connected between the subscriber telephone (10) and the local central office (14), the subscriber interface unit (12) having access to stored data including rules for routing different kinds of calls through the conventional PSTN or through the data network, and a called party interface unit (32; 12A'; 18C) connected to the data network exit node, the method comprising the steps of: at the subscriber interface unit (12; 12A),

- 20 (i) detecting digits dialled by the subscriber to set up a call to said called party;
- (ii) accessing the stored data to determine whether the call should be routed through the Public Switched Telephone Network (PSTN) or through the data network;
- (iii) if it is determined in step (ii) that the call should be routed through the PSTN,
- 25 routing the call to a local central office for processing as a conventional PSTN telephone call;
- (iv) if it is determined in step (ii) that the call should be routed through the data network, connecting to the call server and supplying subscriber interface unit and called party information to the call server;
- 30 at the call server (38),

- (v) using the called party information, identifying and setting up a connection to the called party interface unit (32;12A';18C)  
at the exit network node (30),
- (vii) establishing a connection for completion of the call via the second interface  
5 unit to the called party;  
at the call server (38),
- (viii) notifying the subscriber interface unit (12;12A) that the connection has been established;  
at the subscriber interface unit (12;12A),
- 10 (ix) on receipt of confirmation that the connection to the called party has been completed, converting subsequent signals from the subscriber telephone format to a format compatible with the data network and routing the data signal via the entrance and exit network nodes to the second interface unit;  
at the second interface unit (32;12A';18C),
- 15 (x) converting the data signal into a signal with a format compatible with the called party equipment and routing the compatible signal thereto,
- (xi) converting return signals from the called party equipment into data signals and routing them via the exit and entrance network nodes to the subscriber interface unit (12;12A); and
- 20 at the subscriber interface unit (12;12A),
- (xii) converting the return data signals into the format compatible with the subscriber telephone set and routing the converted signals thereto.

According to a second aspect of the present invention, there is provided a telecommunications system comprising the Public Switched Telephone Network and a public  
25 data network, for example the Internet, and means for routing calls between a subscriber telephone set (10) and a called party (18;18B), the system including a local central office (14), a data network entrance node (26), a data network exit point (30), a call server (38) provisioned with data and software for routing of data calls between the data network entrance node and the data network exit point, and a subscriber interface unit (12; 12A)  
30 connected between the subscriber telephone (10) and the local central office (14), the subscriber interface unit (12) having access to stored data including rules for routing different

kinds of calls through the conventional PSTN or through the data network, and a called party interface unit (32; 12A';18C) connected to the data network exit node, wherein:  
the subscriber interface unit (12;12A) comprises,

5 (i) means (46;48) for detecting digits dialled by the subscriber to set up a call to said called party and accessing the stored data to determine whether the call should be routed through the Public Switched Telephone Network (PSTN) or through the data network;

(i)(a) if it is determined that the call should be routed through the PSTN, routing the call to a local central office for processing as a conventional PSTN.  
10 telephone call;

(i)(b) if it is determined that the call should be routed through the data network, connecting to the call server (38) and supplying subscriber interface unit and called party information to the call server;

the call server (38) comprising means for identifying from the called party information the  
15 called party interface unit (32;12A';18C) and setting up a connection thereto;

the exit network node (30) comprising means for establishing a connection for completion of the call via the second interface unit to the called party;

the call server (38) further comprising means for notifying the subscriber interface unit (12;12A) that the connection has been established;

20 the subscriber interface unit (12;12A) further comprising means operable on receipt of confirmation that the connection to the called party has been completed, to convert subsequent signals from the subscriber telephone format to a format compatible with the data network and route the data signal via the entrance and exit network nodes to the second interface unit;

25 the second interface unit (32;12A';18C) further comprising means for converting the data signal into a signal with a format compatible with the called party equipment and routing the compatible signal thereto, converting return signals from the called party equipment into data signals and routing them via the exit and entrance network nodes to the subscriber interface unit (12;12A);

the subscriber interface unit (12;12A) further comprising means for converting the return data signals into the format compatible with the subscriber telephone set and routing the converted signals thereto.

Preferably, where the subscriber interface unit is connected to the network node via  
5 a narrowband connection, the connection to the network node is initiated while the dialled digits are being collected and the routing decision made, so as to reduce waiting time for the subscriber.

According to a third aspect of the present invention, there is provided a subscriber interface unit for use with the method of the first aspect, the subscriber interface unit  
10 (12;12A) having a first port for connection to a subscriber telephone set, a second port for connection to a subscriber line, means for detecting dialled digits received via the first port and using at least some of the dialled digits to access stored data including routing rules for different kinds of calls to determine whether the call should be routed via the PSTN or via a data network and, in the former case, connecting the first port directly to the second port  
15 and, in the latter case, connecting the first port to the second port via conversion means for converting subscriber signals to data signal having a format suitable for routing through the data network and for converting data signals received from the data network into signals having a format compatible with the subscriber telephone set.

In embodiments of any of these three aspects of the invention, if the subscriber  
20 equipment supplies analog signals, the subscriber interface unit may perform analog-to-digital and digital-to-analog conversion of signals before and after the conversion to and from the data signals. Likewise, if the called party equipment is analog, the second interface unit may provide D-to-A and A-to-D conversion in addition to conversion to and from the data signal format.

25 Where the data network uses packet signals, the subscriber interface unit and the second interface unit may comprise means for converting from subscriber or called party signal format to packetized data signals, and vice versa.

The subscriber interface unit may include a conventional modem for converting the data signals into a format suitable for transmission to the local central office as a PSTN  
30 signal. Alternatively, where the subscriber has a high speed connection to a service provider,

the conventional modem could be omitted and means provided for connecting the data conversion means, e.g. codec, to a high speed modem.

In preferred embodiments, the subscriber interface unit is installed at the subscriber's premises between the telephone set and the second interface unit is a PSTN gateway device  
5 connected to the called party equipment via the PSTN. Where the called party also is a subscriber to the routing service, however, the second interface unit could be similar to the subscriber interface unit.

The stored data accessed by the subscriber interface unit may be stored at the subscriber interface unit itself and periodically updated via the data network by a management  
10 application server unit at a remote location.

In preferred embodiments, the subscriber interface unit has a mechanical switch connected between an input port and an output port for connection to the subscriber telephone set and the subscriber line, respectively, the switch being biased to connect the input port and output directly in the event of power failure to the subscriber interface unit.  
15 Where the subscriber has a data connection to the local central office, the subscriber interface unit may have a port for connection to a high speed modem, in which case the conventional internal modem can be omitted.

Preferably, when a call ends, the connection to the first network node is not taken down immediately in case the subscriber decides to make another call immediately. Rather,  
20 when a first call ends, the interface unit will monitor for the subscriber dialling new digits and, when such digits are detected, determine the routing for the new call, setting up a new communication session, if appropriate, with the call server to set up a new connection to the new called party.

According to a preferred embodiment of the invention, there is provided a process  
25 for routing telephone calls including providing dial tone to a standard telephony device, collecting therefrom dialled digits representing a telephone number of a called party and using at least some of the dialled digits to access stored data and determine whether or not to route the call over a public data network; connecting through the PSTN to a data network connection point (ISP), converting signals from the standard telephony device from analog  
30 to digital, converting the digital signals into packet signals with a format acceptable by the data network, routing the call via the data network to a data connection exit point convenient

for a called party's standard telephony device, receiving return digital signals from the called party standard telephony device via the data network, converting the return digital signals to analog and conveying the analog return signals to the subscriber telephony device.

Thus, the process is capable of routing telephony communications from a public switched telephone network (PSTN) over a public data network (Internet) and back to a public switched telephone network (PSTN).

#### BRIEF DESCRIPTION OF THE DRAWINGS:

An embodiment of the invention will now be described by way of example only with reference to the accompanying drawings in which:

Figure 1 is a schematic diagram of a first embodiment of the present invention for routing telephone calls within a public data network;

Figure 2 is a detail schematic view of a narrowband subscriber interface unit shown in Figure 1;

Figure 3 is a flowchart illustrating a first stage in the setting up of a call using the subscriber interface unit;

Figure 4 is a flowchart illustrating a second stage in the setting up of the call;

Figures 5A and 5B are flowchart sections illustrating the ending of the call in different ways;

Figure 6 illustrates a second embodiment of the invention for routing telephone calls through a public data network using a broadband connection;

Figure 7 illustrates a broadband subscriber interface unit used in the system of Figure 6;

Figure 8 illustrates a first part of a sequence of setting up of a call using the broadband subscriber interface unit;

Figure 9 illustrates a second part of the sequence;

Figure 10 is a flowchart illustrating the ending of a call in different ways using the broadband subscriber interface unit; and

Figure 11 illustrates the handling of an inbound call by the broadband subscriber interface unit.



## DETAILED DESCRIPTION OF PREFERRED EMBODIMENTS

Figure 1 illustrates schematically deployment of a narrowband or "dial-up" embodiment of the invention in the public telecommunications system comprising a Public Switched Telephone Network (PSTN) and a public data network, e.g., the Internet. A station apparatus (telephone set) 10 of a caller who is a service subscriber, i.e., who has paid for the service provided according to the invention, is connected by way of a "router/dialler" or subscriber interface unit 12 and subscriber loop 14 to a local central office of the Public Switched Telephone Network (PSTN) 16. The router/dialler unit 12 is installed at the subscriber's premises and is provided (leased or purchased) to enable the subscriber to use the service.

The station apparatus 18 of a third party (the Called Party) is shown connected via a subscriber loop 20 to a remote or destination central office 22 in the PSTN 16. The local or originating central office 14 and the remote or destination central office 16 are interconnected via the PSTN, as represented by broken line 24.

Among other data, the router/dialler unit 12 stores the telephone number of a "HOME" Internet Service Provider (ISP) 26 accessible via the PSTN and data link 28. The HOME ISP 26, which is provisioned for Voice-over-Internet Protocol (VoIP), is shown connected via the Internet to an EXIT ISP 30, also provisioned for VoIP, which is coupled to a PSTN gateway 32 via a data link 34. The gateway 32 is coupled via link 36 to the PSTN and therefore via the PSTN to the destination central office 22. (It should be noted that, in some situations, the HOME ISP 26 and the EXIT ISP 30 could be the same ISP.)

A call server 38 shown connected to the ISP 26 could be local to the ISP or at a remote location and accessed over the Internet or another data network. The call server 38 is provisioned with a dialled-digits store and routing data and software enabling it to route calls to network nodes, including ISPs, in a known manner.

The data stored by router/dialler unit 12 is downloaded to router/dialler unit 12 during an initialization process carried out once it has been installed and connected, and then updated periodically, by a management application 40, namely software conveniently running on a personal computer acting as a server that is located on the premises of the ISP 26, or at a remote location on the network, and able to communicate with the ISP 26 using the usual protocols. The router/dialler unit 12 is programmed with the telephone number of the

ISP 26 and the name of the management application unit 40 (the name being used for routing purposes) and periodically calls the management application unit 40 for updates, identifying itself by means of its own unique identifier (i.e., serial number), like the unique identifier of a mobile telephone. The management application unit 40 will download to the router/dialler unit 12 whatever information needs to be stored in the router/dialler 12 to enable it to determine whether or not a particular call should be a VoIP call or a regular PSTN call. To this end, the management application 40 is provisioned with feature software enabling it to provide for Administration, Account Verification, Dialing Rule assignment/update and Router/dialler Setup.

10       The typical data that are downloaded by the management application unit 40 will include Dialling rules specifying that some calls are candidates for routing through the data network (VoIP calls) while others should be routed through the PSTN only. For example, international calls or other toll calls might be candidates for VoIP, whereas toll-free calls can be routed as PSTN-only calls. The actual selection would be at the discretion of the service  
15       provider, who might, for example, prefer to include toll-free calls as candidates for VoIP as well. Likewise, if local calls are free, as in most parts of North America, they too may be routed as PSTN-only calls.

When the subscriber begins to make a call, the router/dialler unit 12 will detect the dialled digits, consult the stored data and determine whether the call should be routed as a  
20       regular POTS call via the PSTN or as a Voice-Over-Internet-Protocol (VoIP) call, and route the call accordingly. This happens without the subscriber being involved in the decision.

If the router/dialler unit 12 determines that the call should be routed as a POTS call, it will simply route it via the central office 14 into the PSTN in the usual way (as represented by line 24).

25       If the router/dialler unit 12 determines that the call should be routed as a Voice-over-Internet-Protocol (VoIP) call, the router/dialler converts it into a data signal with the usual header etc. and routes it to the Internet in the usual way via HOME ISP 26, having first set up the connections by exchanging SIP messages with the call server 38. When setting up, the call server 38 detects the dialled digits in a "SIP Invite" message from the router/dialler  
30       12 and uses them to determine the closest gateway network node for the destination, then establishes a connection to PSTN gateway node 32. Once the connection is established, the

gateway 32 supplies the gateway parameters, e.g. compression, CODEC type, and so on, to the router/dialler 12. At the gateway node 32, the call is converted into a PSTN (analog voice) call again and routed via the PSTN to the destination central office 22, which completes the call in the usual way to the called party 18.

5 Referring now to Figure 2, the router/dialler 12 comprises a first port 42 to which the subscriber's telephone set 10 is connected, a mechanical relay 44, a microprocessor 46, a DSP/CODEC 48, a Hayes-compatible modem 50 (specifically a V90 56 kb/s modem), an PSTN port 52 which is connected to the subscriber line, a voltage regulator unit 54 connected to DC input socket 56 for connection to a separate power supply (not shown in  
10 Figure 2) and a Subscriber Line Interface Circuit (SLIC) 58. The voltage regulator 54 supplies the various components at the appropriate voltage. The SLIC 58 provides the subscriber telephone 10 with line voltage etc. which normally would be provided by the central office 14. The DSP/CODEC 48 provides digital-to-analog and analog-to-digital conversion, together with voice compression and decompression to ensure that the data  
15 signals are within the capability of the modem 50. The processor 46 controls the relay 44 and other components, as necessary, and controls the setting up and taking down of PPP communications with the HOME ISP 26 and SIP communications with the call server 38. In addition, the processor 46 converts the digitized voice signals to data signals, specifically VoIP signals, and vice versa.

20 Within the router/dialler 12, the subscriber's telephone set 10 is connected via input port 42 and mechanical relay 44 either to code 48 or directly to output port 52. If there is a power failure, or fault, the relay 44, by default, connects the telephone set 10 to the output port 52 via line 61 so the subscriber can make conventional calls, especially emergency calls, without involving the modem, etc. Moreover, prior to initialization of the recently-installed  
25 router/dialler unit 12, the processor 46 sets the relay 44 to connect the subscriber set directly to the output port 52.

Upon installation, and until the initialization procedure has been completed, all calls are placed to the PSTN. The initialization procedure is triggered when the subscriber lifts the receiver of the phone device 10 and dials a predetermined initialization sequence. This action  
30 will initiate a VoIP connection setup with a call center representative. The call is then terminated at the call center and a customer service representative verifies all applicable

customer profile information with the end user. Once complete, the customer service representative initiates a procedure that provision is the management application server with appropriate dialing rules and the call server with authentication data (if required). The management application then supplies the dialing rules, address of call server, configuration data, and so on, to the router/dialler 12 using a proprietary protocol. The received information is stored in the permanent unit memory and updated periodically by the management application, as necessary. Once the provisioning of the router/dialler unit 12 is complete, it is ready for normal operation.

Assuming normal operating conditions with power supplied to the initialized router/dialler unit 12, the processor 46 will set the mechanical relay 44 to couple the telephone set 10 to the DSP/CODEC 48, i.e., the default presumes that any call will be VoIP. The processor 46 monitors the input port 42 so that, as soon as the subscriber picks up the handset and starts to dial, the processor 46 can detect the "off hook" condition.

Referring now to Figure 3, in which the activities by the subscriber are shown on the right and activities by the processor 46 are shown on the left, in step 3.01, the SLIC 58 is detects the subscriber telephone set "going off-hook" in step 3.02 when the subscriber begins to make the call and reports the event to the processor 46 by way of the DSP/CODEC 48. The setting up of a data call via an ISP takes a significant length of time, so the processor 46 does not wait for the subscriber to dial the called party's telephone before calling the HOME ISP 26. Rather, immediately the processor 46 receives the "off-hook" condition, in step 3.03 the processor 46 activates the modem 50 and, in step 3.04, detects via the modem 50 the usual dial tone supplied by the central office 14. In step 3.05, the processor 46 accesses its stored data to obtain the telephone number of the HOME ISP 26 and dials it via modem 50.

It should be noted that the processor 46 has not yet connected the subscriber to the central office 14 so the subscriber does not hear the usual central office dial tone. Instead, in step 3.06, the processor 46 causes DSP/CODEC 48 to generate a "placebo" dial tone and supply it via the SLIC 58 to the subscriber's telephone set 10. On hearing the placebo dial tone (in step 3.07), the subscriber dials the telephone number of the called party 18 (step 3.08). In step 3.09, the processor 46 collects the dialled digits, instructing the DSP/CODEC 48 to discontinue the placebo dial tone on receipt of the first digit, and in step 3.10 stores the digits temporarily in a buffer. Meanwhile, in step 3.11, the processor 46 is completing the

usual "handshake" process with the HOME ISP 26, supplying login identification and password as appropriate.

In decision step 3.12, the processor 46 accesses its stored data, specifically its "call routing rules", to determine whether the dialled digits represent a destination number to which the call should be routed as a PSTN-only call or as a VoIP call. If the rules specify that the call should be a PSTN-only call, in step 3.13 the processor 46 aborts the establishment of the connection to the HOME ISP 26, or terminates it if completed, then, in step 3.14, retrieves the dialled digits from its buffer and relays them via modem 50 to the central office 14. In step 3.15, the processor 46 switches relay 44 to connect the subscriber's telephone set 10 to line 61 and port 52, thereby connecting the subscriber directly to the central office 14 to hear the usual central office ringing and other call progress tones (step 3.16). The call then continues as a regular PSTN-only call as indicated by step 3.17. If, in step 3.18, either party terminates the call, step 3.19 returns the processor 46 to scanning for originations (step 3.01).

If decision step 3.12 determines that the call should be routed as a VoIP call, the router/dialler performs the functions illustrated in Figure 4. As shown in step 4.01, the processor 46 causes DSP/CODEC 48 to generate an "On Internet" tone and in step 4.02 the SLIC 58 supplies it to the subscriber telephone set 10. In step 4.03, the processor 46 accesses its stored data to determine the network name (address) of call server 38 and, using Session Initiation Protocol (SIP), initiates a call to it by sending it a "SIP Invite" message containing its unique identifier and the dialled digits. On receipt, the call server 38, in step 4.04, accesses its call server authentication data to confirm that the router/dialler 12 is used legitimately.

The call server authentication data is updated periodically by the management application 40 (Figure.1) with up to date authentication data. In step 4.05, the call server 38 uses its routing data to identify the appropriate gateway based upon this routing information. The routing information could be provided either by network administration (static routes) or automatic gateway location services (gatekeepers, ENUM and so on).

In step 4.06, the call server 38 establishes SIP-based communications with the PSTN gateway 32. In step 4.07, the PSTN gateway 32 completes the call to the called party 18 and notifies the call server 38 which supplies a valid termination message to the processor 46.

RTP then is used to set up a bearer connection between the router/dialler 12 and the gateway 32 and ringing tone from the destination central office 22 is conveyed to the processor 46. In step 4.08, the processor 46 conveys the ringing tone via the DSP/CODEC 48 and SLIC 58 to the subscriber telephone set 10 (step 4.09).

5        When, in step 4.10, the called party answers, the call server 38 notifies the processor 46 which, in step 4.11, causes the DSP/CODEC 48 to discontinue the ringing tone. In step 4.12, the DSP/CODEC 48 begins A-D conversion and compression of the voice signal from the subscriber set 10 and the processor 46 converts the compressed digital signal to VoIP data for transmission by the modem 50 to the gateway 32. At the gateway 32, the VoIP data  
10        will be converted to PSTN format (step 4.13) and routed via the PSTN to the called party in the usual way.

Conversely, the PSTN gateway 32 digitizes the called party voice signals and converts them to VoIP sends them via the ISPs 30 and 26 to the router/dialler unit 12 where the processor 46 converts them from VoIP to digital and the DSP/CODEC 48 converts the  
15        digital signals to analog signal and routes the analog signal to the subscriber telephone set 10 via the relay 44. The call proceeds in this manner as indicated in step 4.14, and the processor 46 and gateway 32 monitor for one party or the other ending the call by "going on-hook".

Referring now to Figure 5A, step 5.01 constitutes one of the parties ending the call by "going on-hook" (this is the PSTN call on-hook). If it is the called party 18 who  
20        terminates the call and goes "on-hook" first, in step 5.02 the gateway 32 recognizes the "on-hook" condition and by a SIP message notifies the call server 32 which, on receipt of the notification in step 5.03 relays it to the processor 46 in router/dialler 12. On receipt of the notification in step 5.04 the processor 46 causes DSP/CODEC 48, in step 5.05, to provide a placebo dial tone to the subscriber telephone set 10. The SIP-based VoIP call has now  
25        been taken down.

The processor 46 does not immediately terminate the connection to the HOME ISP 26 but rather, in step 5.06, sets a timer and waits for a predetermined time while monitoring for (i) timer "time-out"; (ii) new dialled digits received, indicating that the subscriber has decided to make another call immediately, without hanging up the handset; and (iii) "off-  
30        hook" detected, to be explained later. If the timer times out, in step 5.07 the processor 46 will cause the DSP/CODEC 48 to discontinue the placebo dial tone and then, in step 5.08,

will "tear down" the connection to the ISP, in step 5.09 wait for the subscriber to hang up and then proceed to step 3.01 (Figure 3) to wait for originations. Discontinuing the placebo dial tone will force the subscriber to hang up and go off-hook again should the subscriber wish to make another call.

5        If the subscriber, on hearing the placebo dial tone in step 5.10, decides to make another call and begins to dial a new telephone number before the timer times out, decision step 5.11 will take the path of "new digits dialled", in which case, in step 5.12, the processor 46 will cancel the timer, in step 5.13 cause the DSP/CODEC 48 to discontinue the placebo dial tone and then proceed to step 3.09 (Figure 3) to collect digits, then continue as before  
10 to set up the new call. In this case, the processor 46 will continue to maintain the connection to the ISP 26.

      If the subscriber does not dial new digits but hangs up instead, from decision step 5.10 the processor 46 proceeds to cancel the timer in step 5.14 and then go to steps 5.07 to 5.09, causing the DSP/CODEC48 to discontinue the placebo dial tone, tearing down the  
15 connection to the HOME ISP 26 and then proceeding to step 3.01 (Figure 3) to scan for originations.

      Referring now to Figure 5B, if it is the caller/subscriber who hangs up first, in step 5.15 the processor 46 detects the "on-hook" condition and in step 5.16 notifies the call server 38 using SIP. On receipt of the notification, in step 5.17, the call server 39 will tear down  
20 the SIP connections to both the processor 46 and the gateway 32. The processor 46 does not terminate the connection to the HOME ISP 26 immediately, however, in case the subscriber decides to make another call immediately. Instead, the processor 46 goes to decision step 5.18, sets the timer and waits, while monitoring for the subscriber to go "off hook" again immediately and start dialling new digits or for the time-out to occur.

25        If no new "off hook" condition is detected before the timer times out, in step 5.19 the processor 46 terminates the connection to the HOME ISP 26, then returns to step 3.01 of Figure 3, i.e., resumes scanning for originations. On the other hand, if an "off-hook" condition is detected before the timer times out, decision step 5.18 takes the "off-hook detected" path, in which case, in step 5.20, the processor 46 will maintain the connection to  
30 the HOME ISP 26 and cancel the outstanding timer, then go to step 3.06 (Figure 3) to cause

DSP/CODEC 48 to supply the placebo dial tone while the processor 46 collects dialled digits in step 3.09 and so on.

If desired for administration and billing purposes, the call server 38 will record when various events occur.

5        Operation of the router/dialler 12 to handle an incoming call is relatively straightforward. When the router/dialler 12 is in its "wait" condition, the processor 46 is monitoring not only the subscriber set 10 via the DSP/CODEC 48 and SLIC 50 for initiation of an outgoing call but also, via modem 50, the line 61 for indications of an incoming call. If the processor 46 detects an incoming call, it merely switches the relay 44 to connect line 61  
10 directly to the subscriber's set 10. The processor 46 monitors line 61 to detect completion of the call at which point it causes relay 44 to return to its original state.

Referring again to Figure 1, if the call is destined for a called party 18B who has a similar broadband router/dialler unit 12A', the latter can communicate directly with the call server 38, i.e., without going through gateway 32, and itself convert from VOIP to analog.  
15 Thus, in this case, the codec units 48 in the two router/diallers 12A and 12A' will handle the A-to-D conversion and compression, and vice versa, as appropriate. It would also be possible to complete the call to a called party with a SIP telephone or other compatible device elsewhere on the Internet, as indicated by optional ON NET EXTERNAL box 67.

A second embodiment of the invention, which used a broadband router/dialler, will  
20 now be described with reference to Figures 6 to 12. Components of the broadband system shown in Figure 6 which are the same as component in the embodiment of Figures 1 and 2 have the same reference numerals. Referring first to Figure 6, the subscriber telephone set 10 is connected to a broadband router/dialler 12A which differs slightly from the narrowband router/dialler 12 of Figures 1 and 2. As shown in Figure 7, the router/dialler 12A has three  
25 ports, a first port 42 for connection to the subscriber telephone set 10, a second port 52 for connection to the regular telephone line for transmission of PSTN-only analog telephone calls, and a third port 62 for connection via a high speed modem 63 (see Figure 6) to a broadband subscriber line.

The router/dialler 12A also differs from router/dialler 12 in that it has no Hayes-  
30 compatible modem; instead it has a Data Access Arrangement (DAA) unit 64 connected between the processor 46 and the output port 52 and a DTMF transceiver 65 connected



between the processor 46 and the DAA 63. The DAA 64 interfaces with the central office 14 and provides various functions normally provided by the subscriber set 10, such as detecting voltage changes indicating that the central office 14 is "off-hook", detecting ringing tones, and so on. The DTMF transceiver 65 detects ringing tones from the central office and provides a means whereby the processor 46 can dial digits for transmission to the central office 14.

In Figure 6, port 52 is shown connected via subscriber loop 66 to the central office 14 of the PSTN 16 and port 62 is shown connected via high speed modem 63 to HOME ISP 26. It will be appreciated that, if the broadband connection is an Asynchronous Digital Subscriber Line (ADSL), both the high speed modem 63 and the regular PSTN port 52 would be connected via the same subscriber loop 66 to the central office 14 where a digital subscriber loop access multiplexer (DSLAM) would separate the PSTN-only calls and the data calls and route them to the PSTN 16 and the HOME ISP 26, respectively.

The high speed or data connection could, of course, be by telephone line, cable or a wireless/satellite connection. In all cases, however, it is presumed that the router/dialler 12A does not need to set up a call to the HOME ISP 26 because it is in continuous communication with it.

Operation of the system of Figure 6 for different calls will now be described with reference first to Figure 8. In step 8.01, the processor 46 scans the subscriber telephone port 42 and line 61 for originations. If, in step 8.02, the subscriber picks up the handset to make a call, in step 8.03 the processor 46 causes the DSP/CODEC 48 to supply a placebo dial tone to the subscriber set 10. On hearing the placebo dial tone in step 8.04, the subscriber begins to dial the destination telephone number of the called party 18 (step 8.05). In step 8.06, the processor 46 collects the dialled digits and, in step 8.07, buffers them while determining the routing for the call in step 8.08, using the rules previously stored in the router/dialler 12A. As in the case of the narrowband router/dialler 12, the management application 40 downloads to the router/dialler 12A dialling rules and other data and updates it periodically.

If the processor 46 determines that the call should be routed as a PSTN-only call, in step 8.09 the processor 46 opens a connection to the central office via DTMF transceiver 65 and DAA 64 and transmits to the central office 14 the dialled digits from its buffer. Once the call has been connected, in step 8.11 the processor 46 switches relay 44 to connect the

subscriber set 10 directly to the line 61 and thence to central office 14, whereupon, in step 8.12, the subscriber hears the ringing tone. In step 8.13, the call proceeds as a regular PSTN call until, in step 8.14, either party terminates whereupon, in step 8.15, the processor 46 returns to scanning for originations (step 8.01).

5        If, in step 8.08, the processor 46 determines that the call should be routed as a data (VOIP) call, the processor 46 proceeds to step 9.01 and causes the DSP/CODEC 48 to supply to the subscriber set 10 an "On Internet" tone to be heard by the subscriber (step 9.02). In step 9.03, processor 46 accesses its stored data to determine the network name/address of the call server 38 and uses SIP protocol to initiate a call to the call server 38, supplying its own  
10 unique identifier and the destination number. In step 9.04, the call server 38 accesses its authentication data to confirm that the router/dialler 12A is used legitimately. In step 9.05, the call server 38 uses its routing data to determine the network name/address of the appropriate PSTN gateway 32 and in step 9.06 establish a SIP-based connection to the gateway 32 which completes the connection via the PSTN to the called party 18.

15        In step 9.07, the destination central office 22 supplies the usual valid termination message which the gateway 32 relays, using SIP, to the call server 38 and the processor 46. On receipt of the valid termination message, in step 9.08 the processor 46 causes the DSP/CODEC 48 to supply to the subscriber 10 a ringing tone. In step 9.09, the call server 38 detects that the called party 18 has answered and relays the information to the processor  
20 46 which, in step 9.10, instructs the DSP/CODEC 48 to discontinue the ringing tone and allows the call to proceed.

As shown in step 9.11, the DSP/CODEC 48 digitizes and compresses the analog signal from subscriber set 10 and the processor 46 converts the resulting digital signal to VOIP format and relays the VOIP signals via DAA 64, high speed modem 63, HOME ISP 26,  
25 PSTN gateway 32 and gateway 32 to the called party 18. As before, the gateway 32 converts the VOIP signal to analog form again before routing it to the called party 18.

Conversely, in step 9.12, the gateway 32 digitizes the analog signal from called party 18, converts it to VOIP format and routes it to the router/dialler 12A wherein the processor 46 converts it from VOIP to digital and DSP/CODEC 48 decompresses the digital signal,  
30 converts it to analog form, and conveys it to the subscriber 10.

As described with reference to the narrowband router/dialler 12, the calls from broadband router/dialler 12A also could be made to a subscriber 18B or to a SIP phone 18C or other ON NET EXTERNAL device.

Figure 10 illustrates the sequence of operations when either of the parties ends the call, i.e., "goes on-hook" in step 10.01. If it is the called party 18 who terminates the call and goes "on-hook" first, the gateway 32 detects the event and issues a SIP message to notify the call server 38. In step 10.02, the call server 38 recognizes the "on-hook" condition and, in a similar SIP message, notifies the processor 46 in router/dialler 12A. On receipt of the notification in step 10.03, the processor 46 proceeds to step 8.03 to allow further calls by the subscribers. At this point, the SIP-based connection is down.

If it is the caller/subscriber who hangs up first, in step 10.04 the processor 46 detects the "on-hook" condition and in step 10.05 notifies the call server 38 using SIP. In step 10.06, the call server 38 receives the message to take down the SIP connection and proceeds to do.

so according to SIP procedures. The processor 46 then goes to step 8.01 (Figure 8) to scan for originations.

As before, if desired for administration and billing purposes, the call server 38 will record when various events occur.

5        Operation of the broadband router/dialler 12A to handle an incoming call is relatively straightforward. As shown in Figure 11, when the router/dialler 12A is in its "scant" condition, step 11.01, the processor 46 is monitoring the subscriber set 10 for initiation of an outgoing call while the processor 46, via DAA 64 and DTMF transceiver 65, monitors the line 61 for indications of an incoming call. When the processor 46 and DAA 64 present on "on-  
10 hook" condition, the central office transmits a "ringback" signal to the calling party and then the processor 46 in step 11.03 determines whether the subscriber set 10 is "on-hook". If it is, in step 11.04 the processor 46 connects the call to subscriber set 10 and in step 11.05 the ringing tone is heard by the calling party.

If step 11.03 determines that the subscriber set is "off-hook", in step 11.06 the  
15 processor 46 determines whether or not there is a VOIP call in progress. If there is not, the processor 46 loops back to repeat steps 11.05 and 11.06 to periodically retests both conditions until it is positive or the inbound call is abandoned.

If step 11.06 determines that there is a VOIP call in progress, step 11.07 supplies alert tones for a predetermined period to the calling party. These alert tones, when heard by the  
20 subscriber, inform the subscriber that there is an inbound call waiting. To accept the inbound call, the subscriber must terminate the VOIP call and then accept the inbound call by "going off-hook".

To summarize, in systems embodying the present invention, callers use a Voice-over-IP system to place outgoing calls over an ISP's network. Customers use their own telephone  
25 instrument, plugged into a router/dialler, to access the ISP's network. The router/dialler connects to the ISP's network using a PPP connection when the customer places a call. The ISP operates a number of PSTN Gateways which bridge voice calls between the ISP's network and the PSTN. Calls are processed by a Call Server, which performs authentication, gateway selection and, optionally, call logging. A management application exists to enable  
30 support personnel to configure new router/diallers or update gateway lists.

Although, in the above-described embodiment, the public data network is the Internet, it will be appreciated that, with suitable modification, the invention could work with other data networks.

## 5 INDUSTRIAL APPLICABILITY

Advantageously, embodiments of the present invention provide selective routing of conventional POTS telephone communications via either the PSTN or a public data network without user intervention, facilitating optimization of route selection and transmission over the data network. This may reduce the cost of communication and avoid the need for a PBX-  
10 to-data network switching device.

Embodiments of the invention overcome the main drawbacks of the aforementioned conventional routing systems and provide a system which enables standard telephone sets to communicate with other standard telephone sets by re-routing the communication through a public switched telephone system (PSTN) on to the public data system (Internet) and back  
15 onto the public switched telephone system (PSTN), thereby overcoming the requirement for access to a private data network of previous systems. An important advantage of embodiments of the invention is that no user intervention is required when making the decision whether to route the call via the PSTN only or through interface with the data network. Nor does the user need to have any knowledge of the recipient's IP address.

## CLAIMS:

1. A method of routing telephone calls from a subscriber telephone (10) in a telecommunications system comprising the Public Switched Telephone Network and a public  
5 data network, for example the Internet, to a called party, the system including a local central office (14), a data network entrance node (26), a data network exit point (30), a call server (38) provisioned with data and software for routing of data calls between the data network entrance node and the data network exit point, and a subscriber interface unit (12; 12A) connected between the subscriber telephone (10) and the local central office (14), the  
10 subscriber interface unit (12) having access to stored data including rules for routing different kinds of calls through the conventional PSTN or through the data network, and a called party interface unit (32; 12A'; 18C) connected to the data network exit node, the method comprising the steps of:

at the subscriber interface unit (12; 12A),

- 15 (i) detecting digits dialled by the subscriber to set up a call to said called party;  
(ii) accessing the stored data to determine whether the call should be routed through the Public Switched Telephone Network (PSTN) or through the data network;  
(iii) if it is determined in step (ii) that the call should be routed through the PSTN,  
20 routing the call to a local central office for processing as a conventional PSTN telephone call;  
(iv) if it is determined in step (ii) that the call should be routed through the data network, connecting to the call server and supplying subscriber interface unit and called party information to the call server;

25 at the call server (38),

- (v) using the called party information, identifying and setting up a connection to the called party interface unit (32; 12A'; 18C)

at the exit network node (30),

- (vii) establishing a connection for completion of the call via the second interface  
30 unit to the called party;

at the call server (38),

- (viii) notifying the subscriber interface unit (12;12A) that the connection has been established;
- at the subscriber interface unit (12;12A),
- (ix) on receipt of confirmation that the connection to the called party has been .
- 5 completed, converting subsequent signals from the subscriber telephone format to a format compatible with the data network and routing the data signal via the entrance and exit network nodes to the second interface unit;
- at the second interface unit (32;12A';18C),
- (x) converting the data signal into a signal with a format compatible with the
- 10 called party equipment and routing the compatible signal thereto,
- (xi) converting return signals from the called party equipment into data signals and routing them via the exit and entrance network nodes to the subscriber interface unit (12;12A); and
- at the subscriber interface unit (12;12A),
- 15 (xii) converting the return data signals into the format compatible with the subscriber telephone set and routing the converted signals thereto.
2. A method according to claim 1, wherein the subscriber telephone set (10) supplies analog signals and the subscriber interface unit (12) performs analog-to-digital and digital-to-
- 20 analog conversion of signals before or after the conversion from data network protocol to digital and vice versa.
3. A method according to claim 1 or 2, wherein the called party equipment supplies analog signals and the called party interface unit (32;12A';18C) provides D-to-A and A-to-D
- 25 conversion.
4. A method according to claim 1, 2 or 3, wherein the stored data accessed by the subscriber interface unit (12;12A) is stored at the subscriber interface unit and periodically updated via the data network by a management application unit (40) at a remote location.

5. A method according to claim 1, 2, 3 or 4, wherein the subscriber interface unit is connected to the entrance network node via a narrowband connection, and the connection to the entrance network node is initiated while the dialled digits are being collected and the routing decision made.

5

6. A method according to claim 1, 2, 3 or 4, wherein the subscriber interface unit is connected to the entrance network node via a narrowband connection and, when a particular call has ended, the subscriber interface unit maintains the connection to the entrance network node for a predetermined time while monitoring for a second call being initiated by the  
10 subscriber.

7. A telecommunications system comprising the Public Switched Telephone Network and a public data network, for example the Internet, and means for routing calls between a subscriber telephone set (10) and a called party (18;18B), the system including a local central  
15 office (14), a data network entrance node (26), a data network exit point (30), a call server (38) provisioned with data and software for routing of data calls between the data network entrance node and the data network exit point, and a subscriber interface unit (12; 12A) connected between the subscriber telephone (10) and the local central office (14), the subscriber interface unit (12) having access to stored data including rules for routing different  
20 kinds of calls through the conventional PSTN or through the data network, and a called party interface unit (32; 12A';18C) connected to the data network exit node, wherein:  
the subscriber interface unit (12;12A) comprises,

(i) means (46;48) for detecting digits dialled by the subscriber to set up a call to said called party and accessing the stored data to determine whether the call should  
25 be routed through the Public Switched Telephone Network (PSTN) or through the data network;

(i)(a) if it is determined that the call should be routed through the PSTN, routing the call to a local central office for processing as a conventional PSTN telephone call;



- (i)(b) if it is determined that the call should be routed through the data network, connecting to the call server (38) and supplying subscriber interface unit and called party information to the call server;
- the call server (38) comprising means for identifying from the called party information the  
5 called party interface unit (32;12A';18C) and setting up a connection thereto;
- the exit network node (30) comprising means for establishing a connection for completion of the call via the second interface unit to the called party;
- the call server (38) further comprising means for notifying the subscriber interface unit (12;12A) that the connection has been established;
- 10 the subscriber interface unit (12;12A) further comprising means operable on receipt of confirmation that the connection to the called party has been completed, to convert subsequent signals from the subscriber telephone format to a format compatible with the data network and route the data signal via the entrance and exit network nodes to the second interface unit;
- 15 the second interface unit (32;12A';18C) further comprising means for converting the data signal into a signal with a format compatible with the called party equipment and routing the compatible signal thereto, converting return signals from the called party equipment into data signals and routing them via the exit and entrance network nodes to the subscriber interface unit (12;12A);
- 20 the subscriber interface unit (12;12A) further comprising means for converting the return data signals into the format compatible with the subscriber telephone set and routing the converted signals thereto.
8. A system according to claim 7, wherein the subscriber telephone set is analog, the  
25 subscriber interface unit (12) further comprising means for converting the subscriber signals from analog to digital before supplying same to the data network and for converting signals from digital to analog before passing same to the subscriber telephone set.

9. A system according to claim 7 or 8, wherein the called party telephone set is analog,  
30 the called party interface unit having D-to-A and A-to-D conversion capability.

10. A system according to claim 7, 8 or 9, wherein the subscriber interface unit has a mechanical switch connected between a first port and a second port for connection to the subscriber telephone set and the subscriber line, respectively, the switch being biased to connect the first and second ports directly in the event of power failure to the subscriber  
5 interface unit.

11. A method according to claim 7, 8, 9 or 10, wherein the subscriber interface unit is connected to the entrance network node via a narrowband connection, and the connection to the entrance network node is initiated while the dialled digits are being collected and the  
10 routing decision made.

12. A method according to claim 7, 8, 9 or 10, wherein the subscriber interface unit is connected to the entrance network node via a narrowband connection and, when a particular call has ended, the subscriber interface unit maintains the connection to the entrance network  
15 node for a predetermined time while monitoring for a second call being initiated by the subscriber.

13. A subscriber interface unit for use with the method of the first aspect, the subscriber interface unit (12;12A) having a first port (42) for connection to a subscriber telephone set  
20 (10), a second port (52) for connection to a subscriber line, means (46,48) for detecting dialled digits received via the first port and using at least some of the dialled digits to access stored data including routing rules for different kinds of calls to determine whether the call should be routed via the PSTN or via a data network and, in the former case, connecting the first port (42) directly to the second port (52) and, in the latter case, converting subscriber  
25 signals to data signals having a format suitable for routing through the data network and for converting data signals received from the data network into signals having a format compatible with the subscriber telephone set, the data signals being routed to and from the second port via modem means (50).

30 14. A subscriber interface unit according to claim 13, for use where the subscriber has a data connection to a network entrance node, the subscriber interface unit having a further

port (62) for connection to a broadband modem (63), and means (64,65) for communicating set-up data signals to and from the PSTN port (52), the data signals being routed to and from the further port (62).

- 5 15. A process for providing dial tone to a standard telephony device, collecting therefrom dialled digits representing a telephone number of a called party and comparing at least some of the dialled digits with stored data to determine whether or not to route the call over a public data network; connecting through the PSTN to a data network connection point (ISP), converting signals from the standard telephony device from analog to digital, placing the  
10 digital signals into packets acceptable by the data network, routing the call via the data network to a data connection exit point convenient for a called party standard telephony device, receiving return digital signals from the called party standard telephony device via the data network, converting the return digital signals to analog and conveying the analog return signals to the subscriber telephony device.

## ABSTRACT OF THE DISCLOSURE

In a method of routing calls in a telecommunications system comprising a PSTN (16) and a public data network, a subscriber interface unit (12) provides dial tone for a standard 5 telephony device (10), collects the dialled digits therefrom and decides whether or not to route the call over the data network; connects through the PSTN to a data network connection point (26), converts the communication from analog to compressed digital form, places the digital communication into VOIP form, and routes the call via the data network to data connection exit point (30) convenient to the called party. The subscriber interface unit 10 converts to analog form the VOIP data signals received from the called party via the data network and conveys same to the subscriber telephony device.

1/12

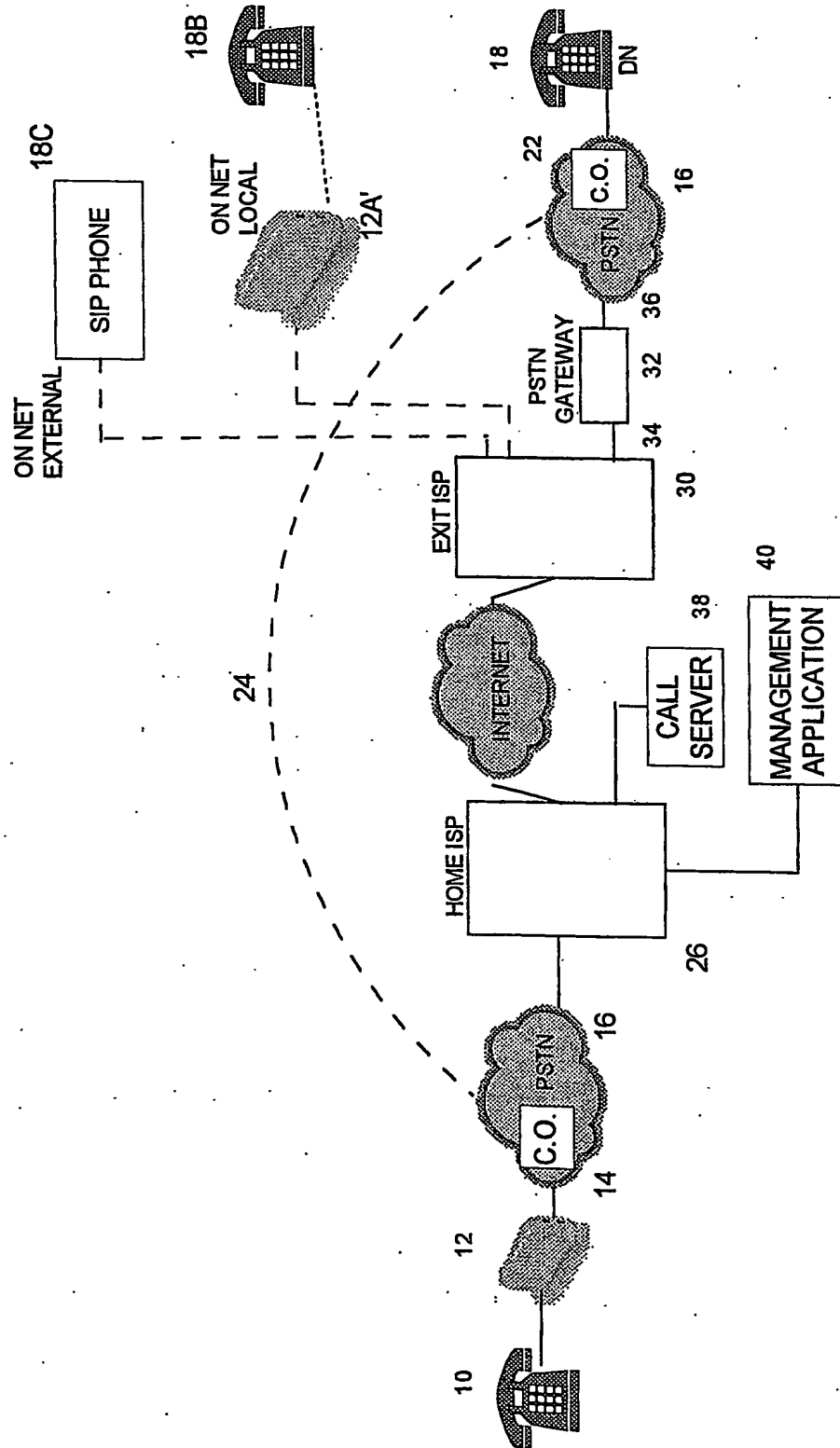


FIG. 1

2/12

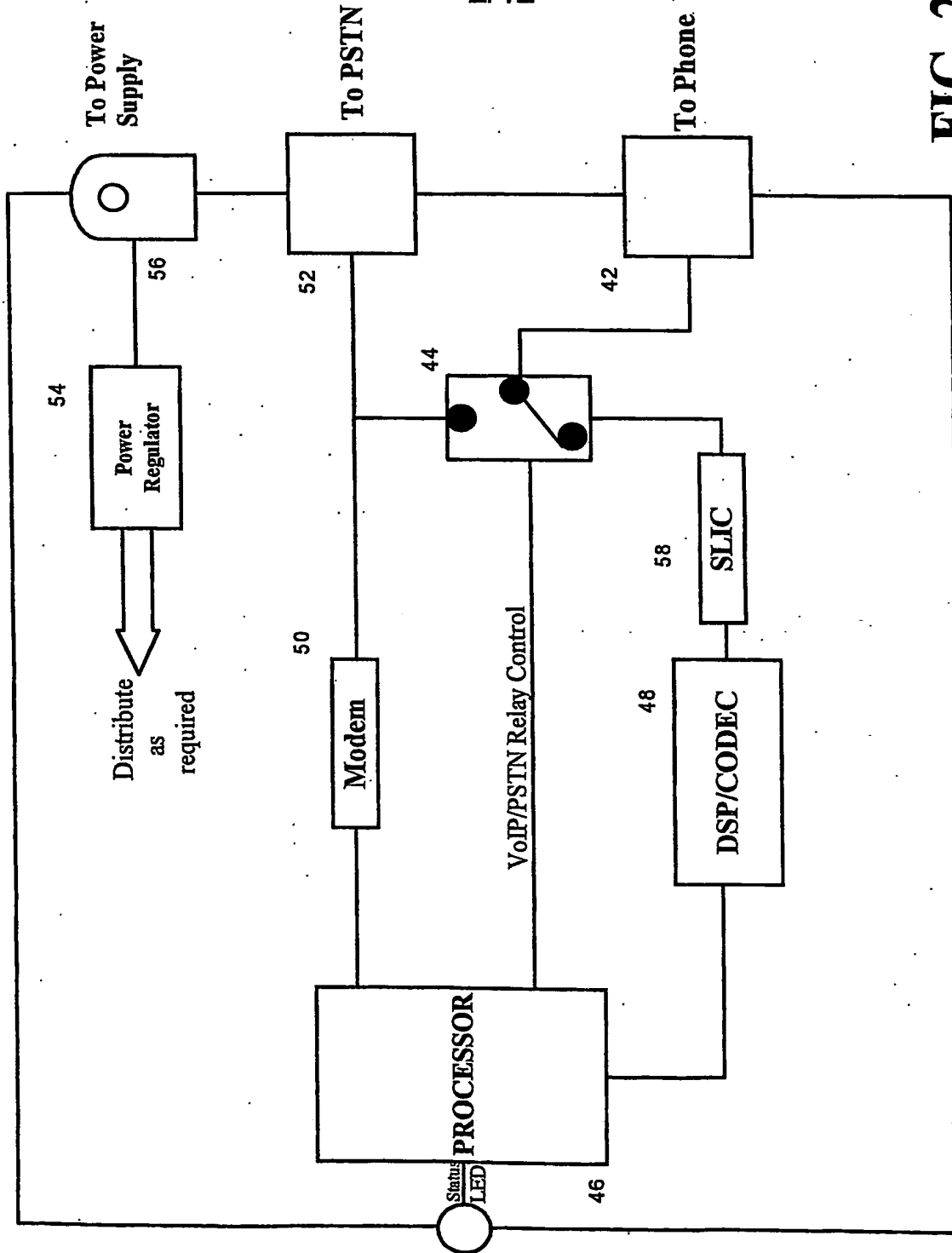


FIG. 2

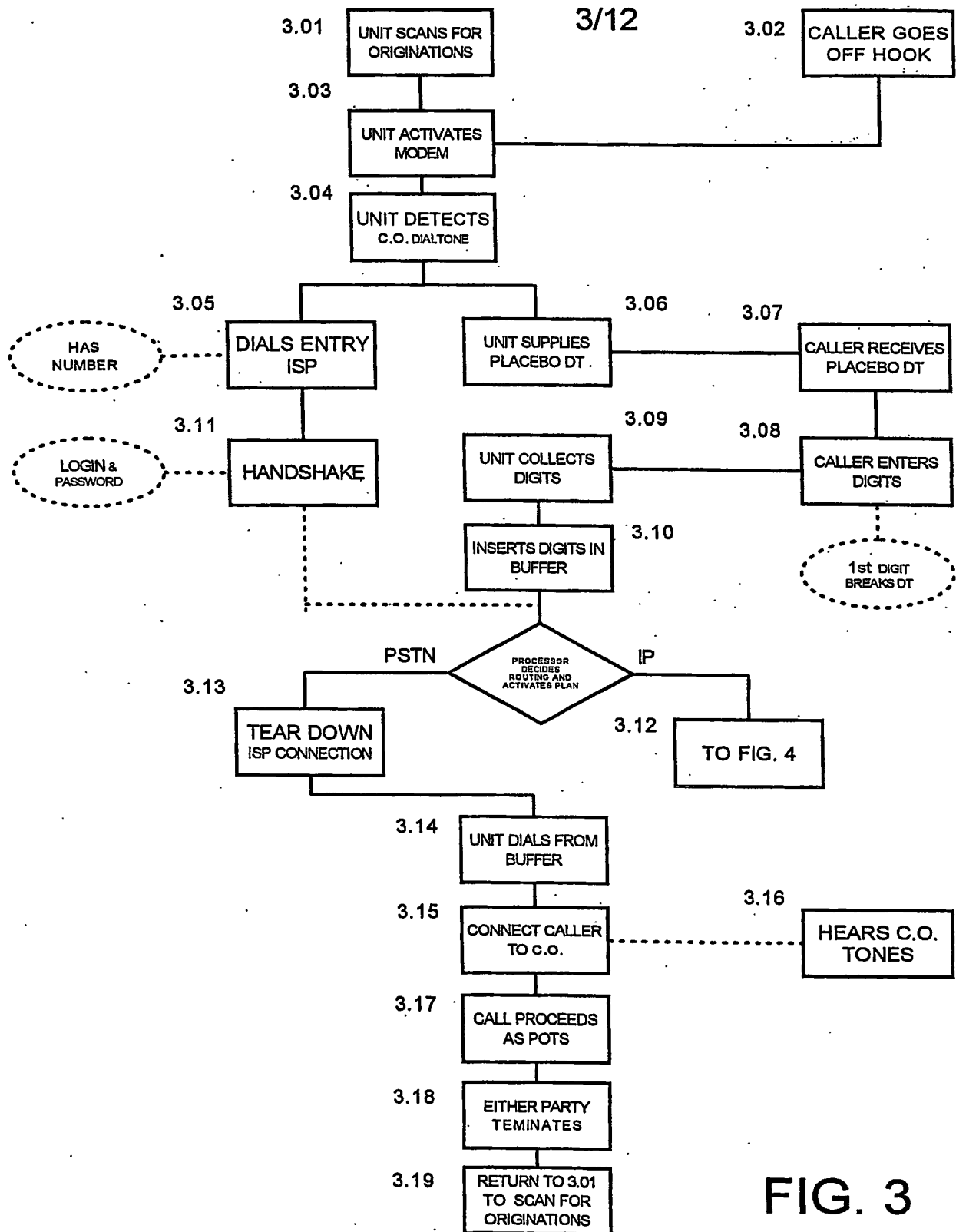


FIG. 3

4/12

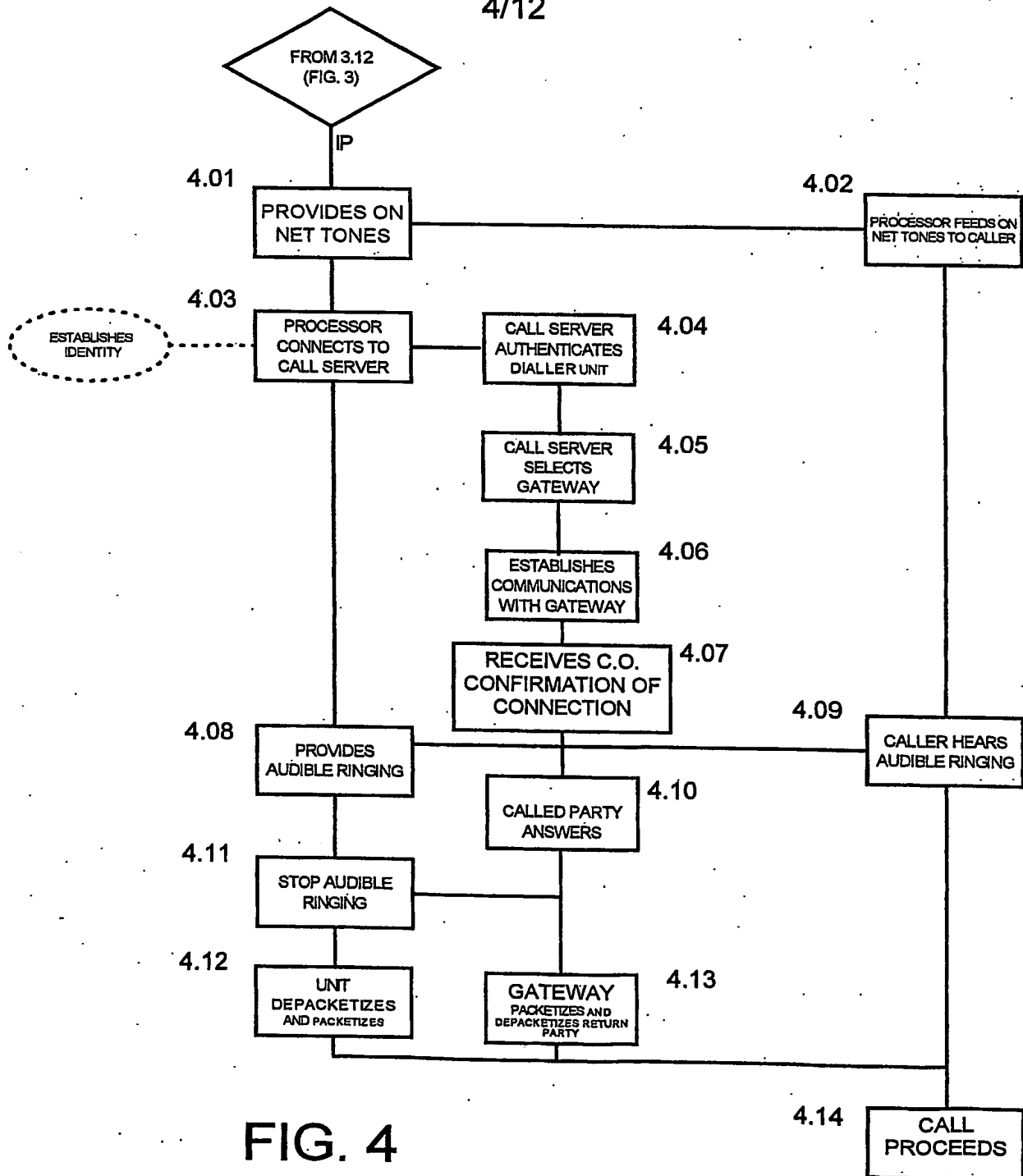


FIG. 4



5/12

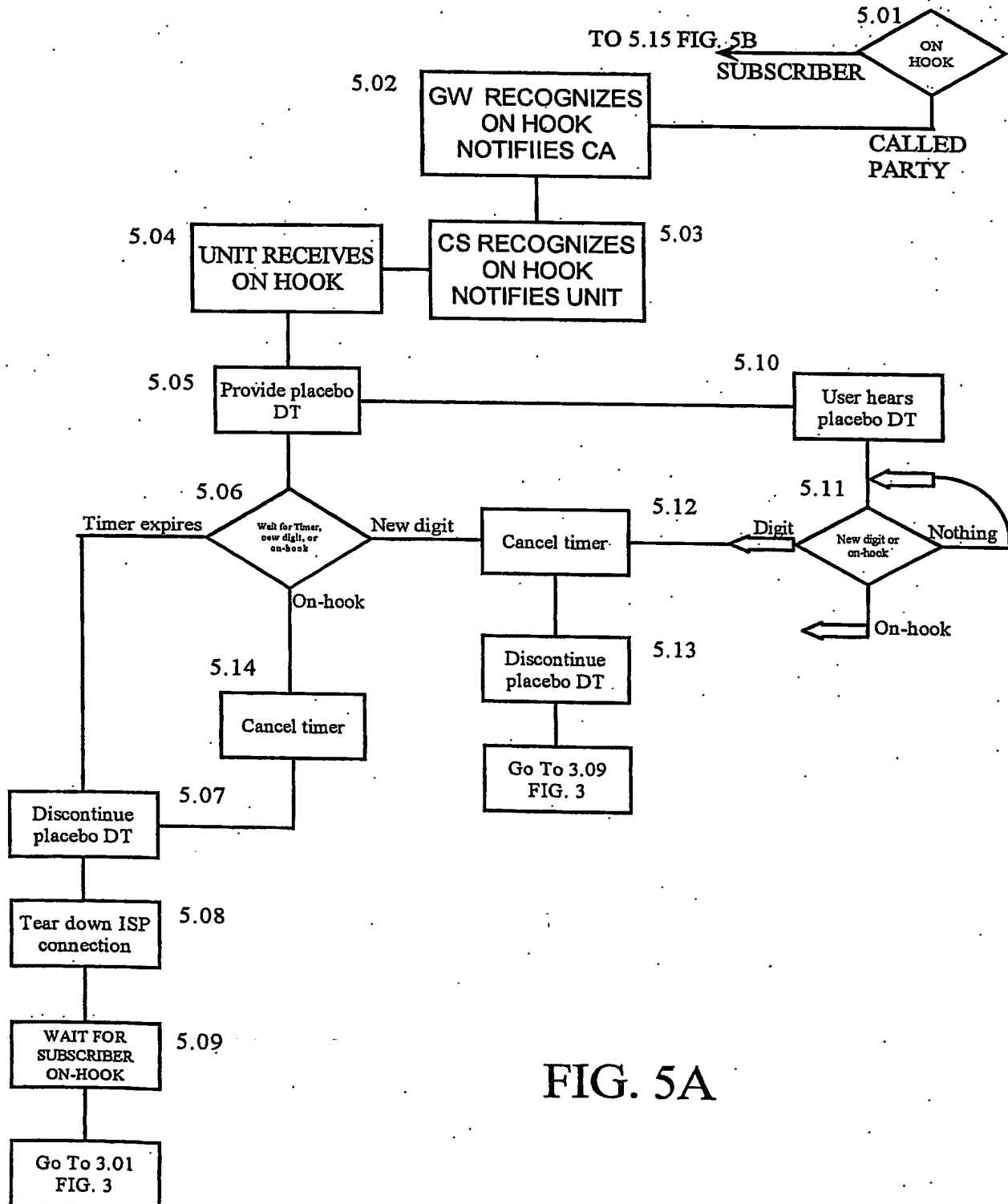


FIG. 5A

6/12

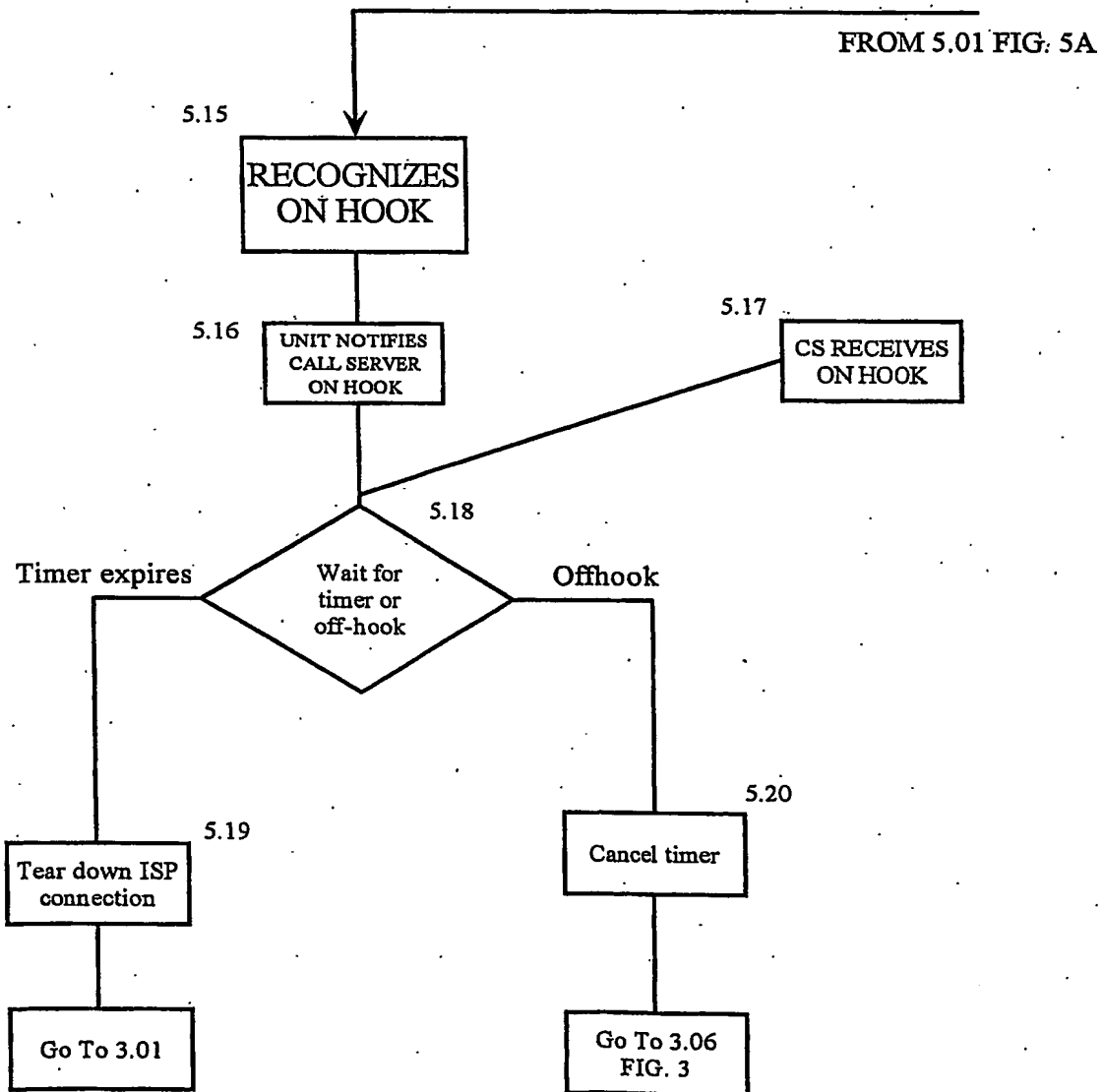


FIG. 5B

7/12

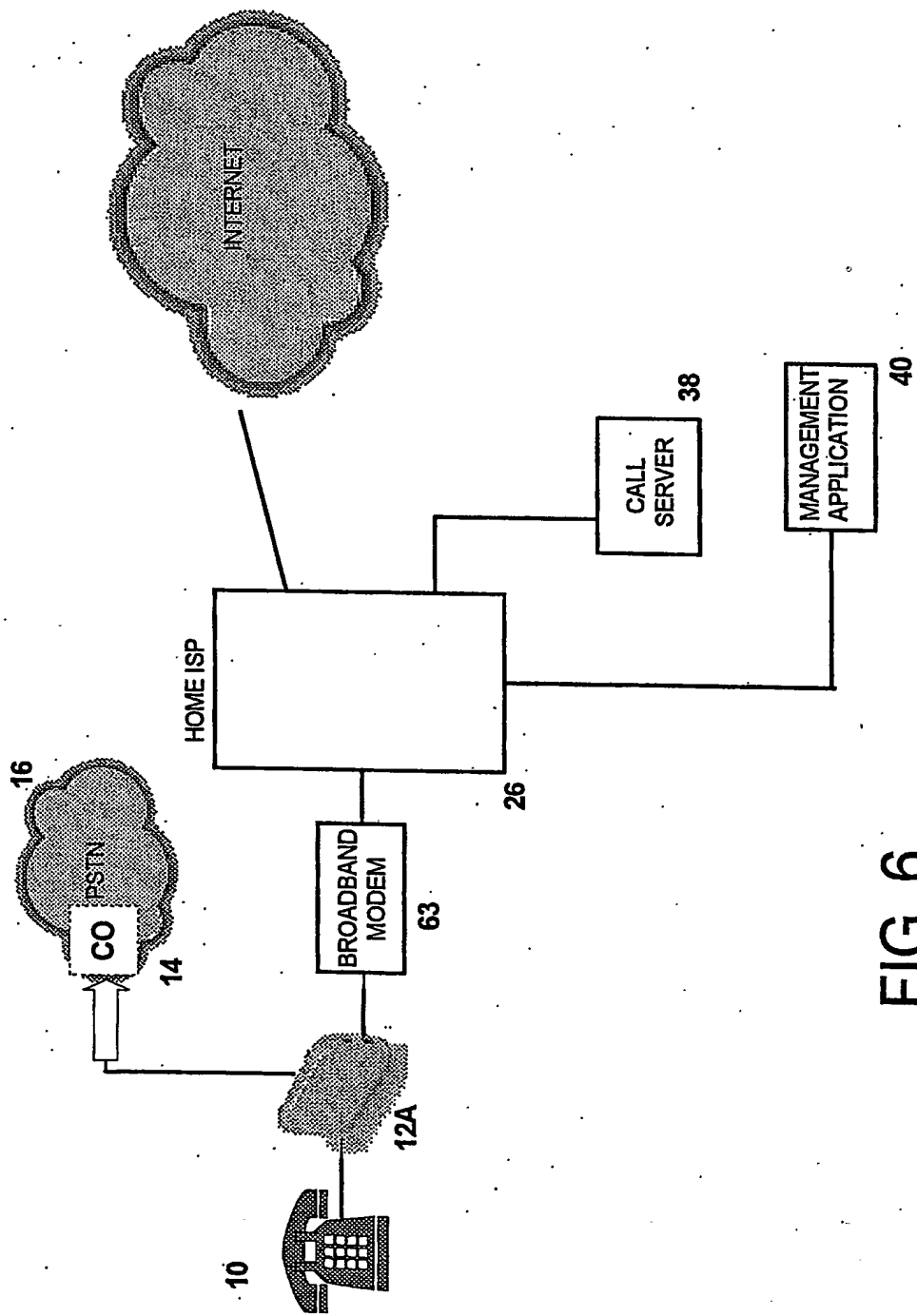


FIG. 6

12A

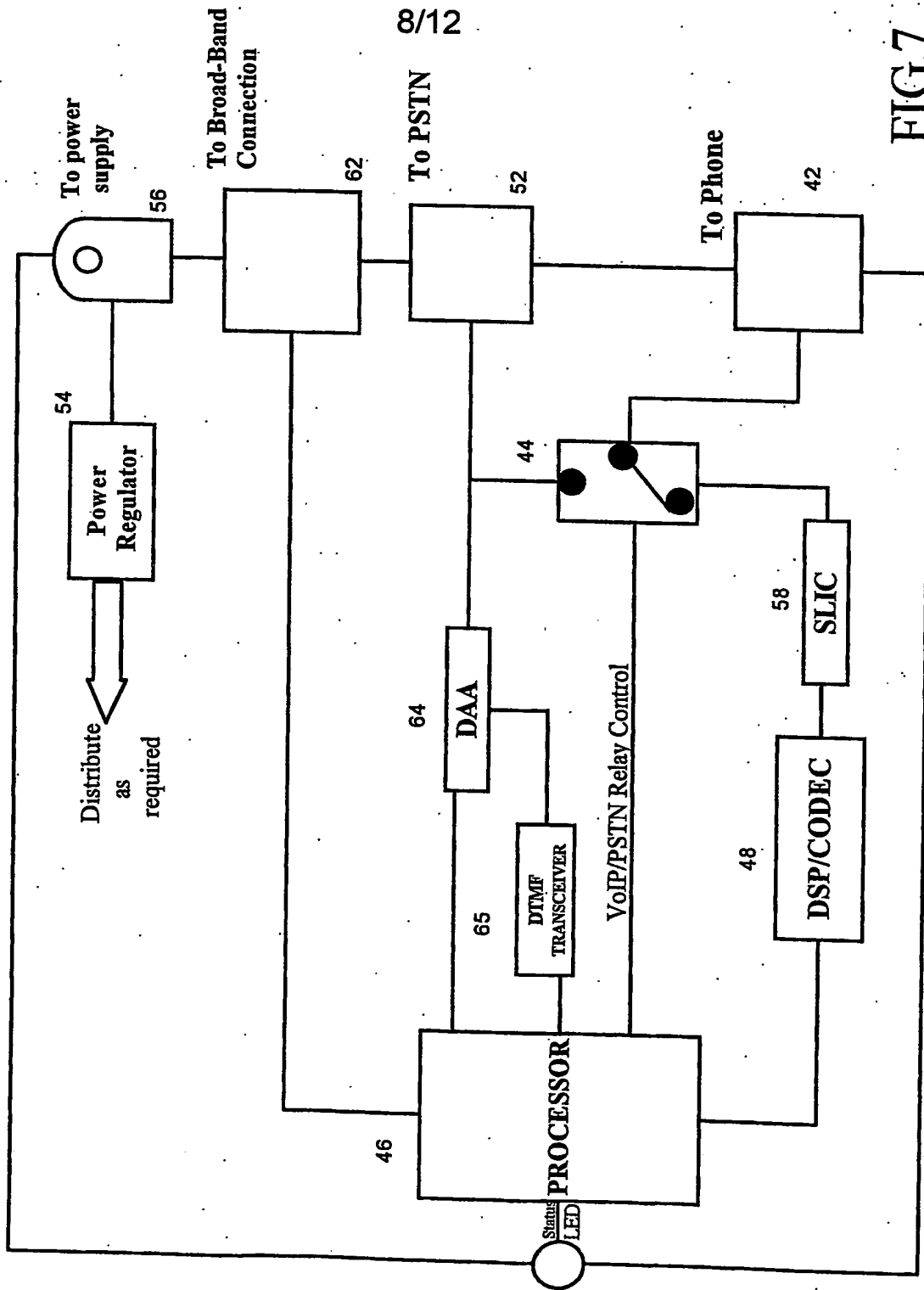
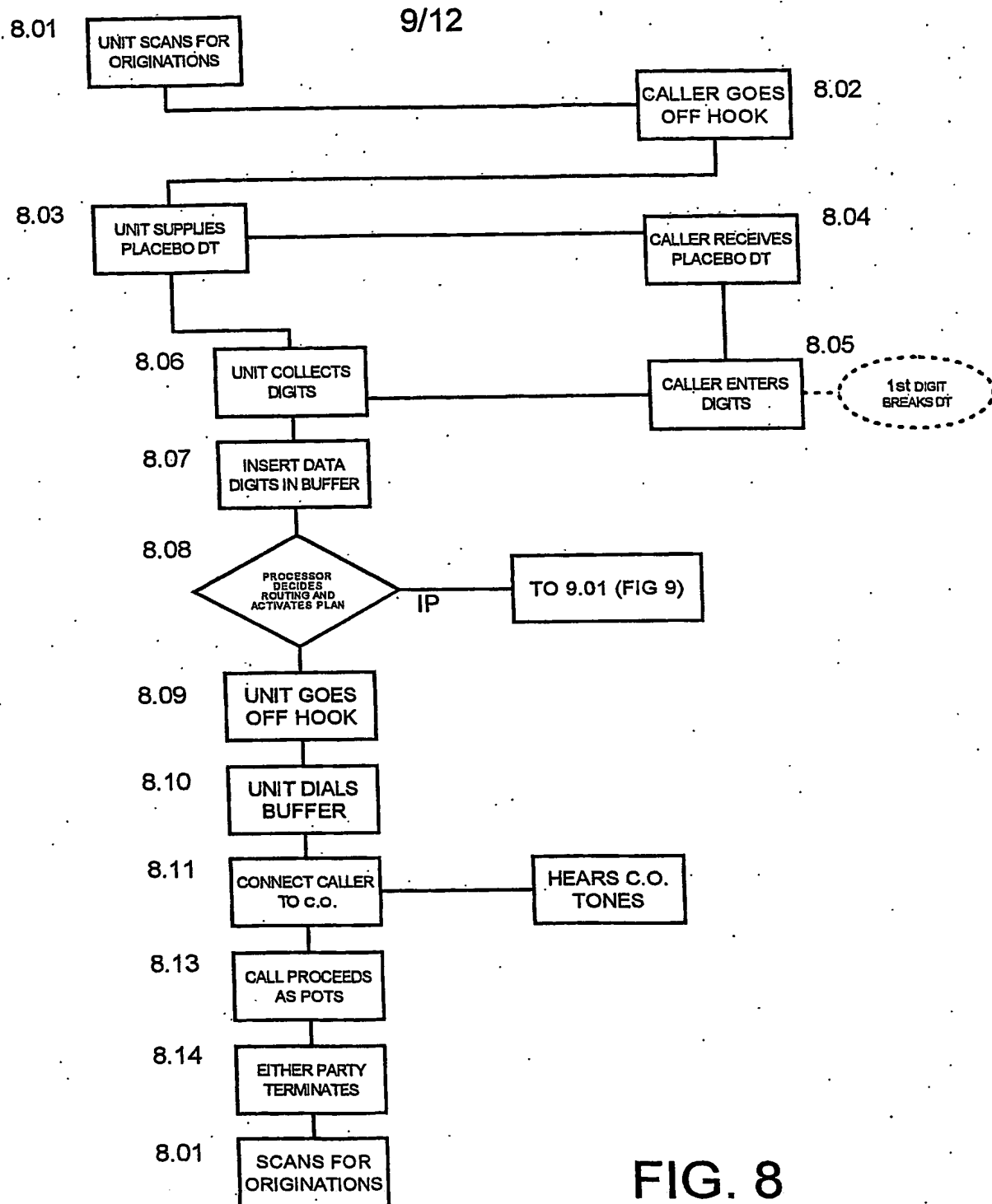


FIG. 7



10/12

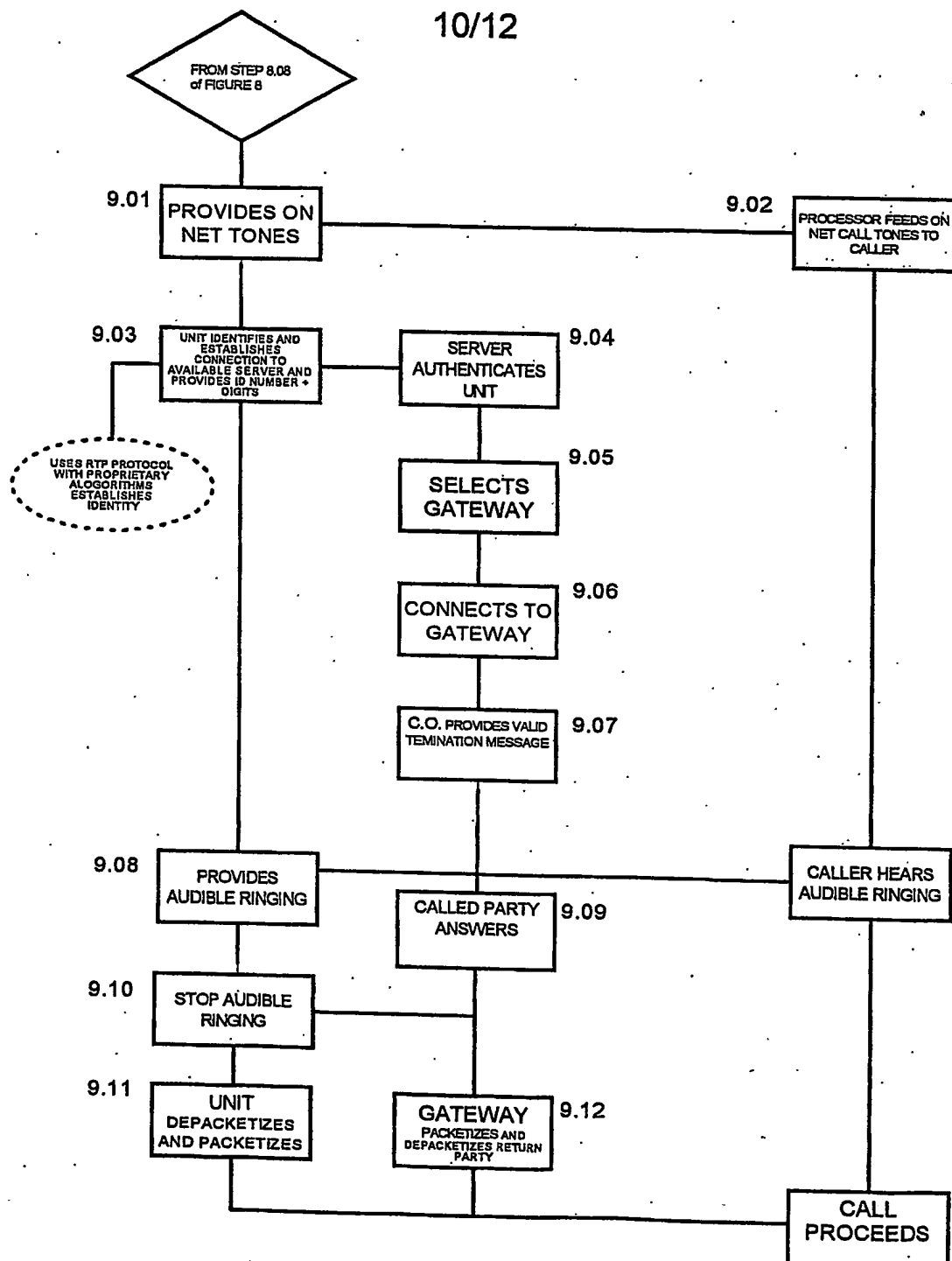


FIG. 9

11/12

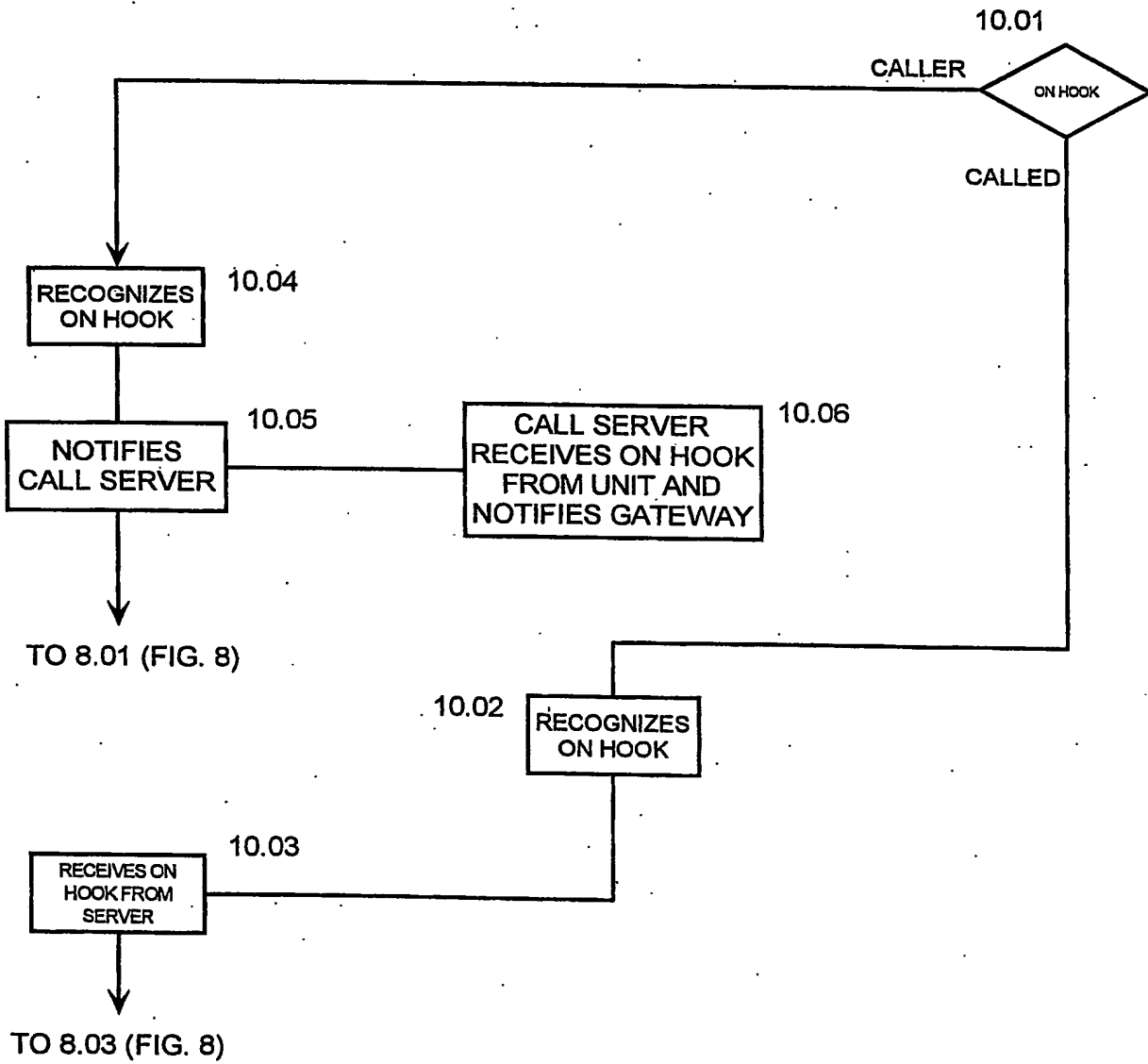


FIG. 10

12/12

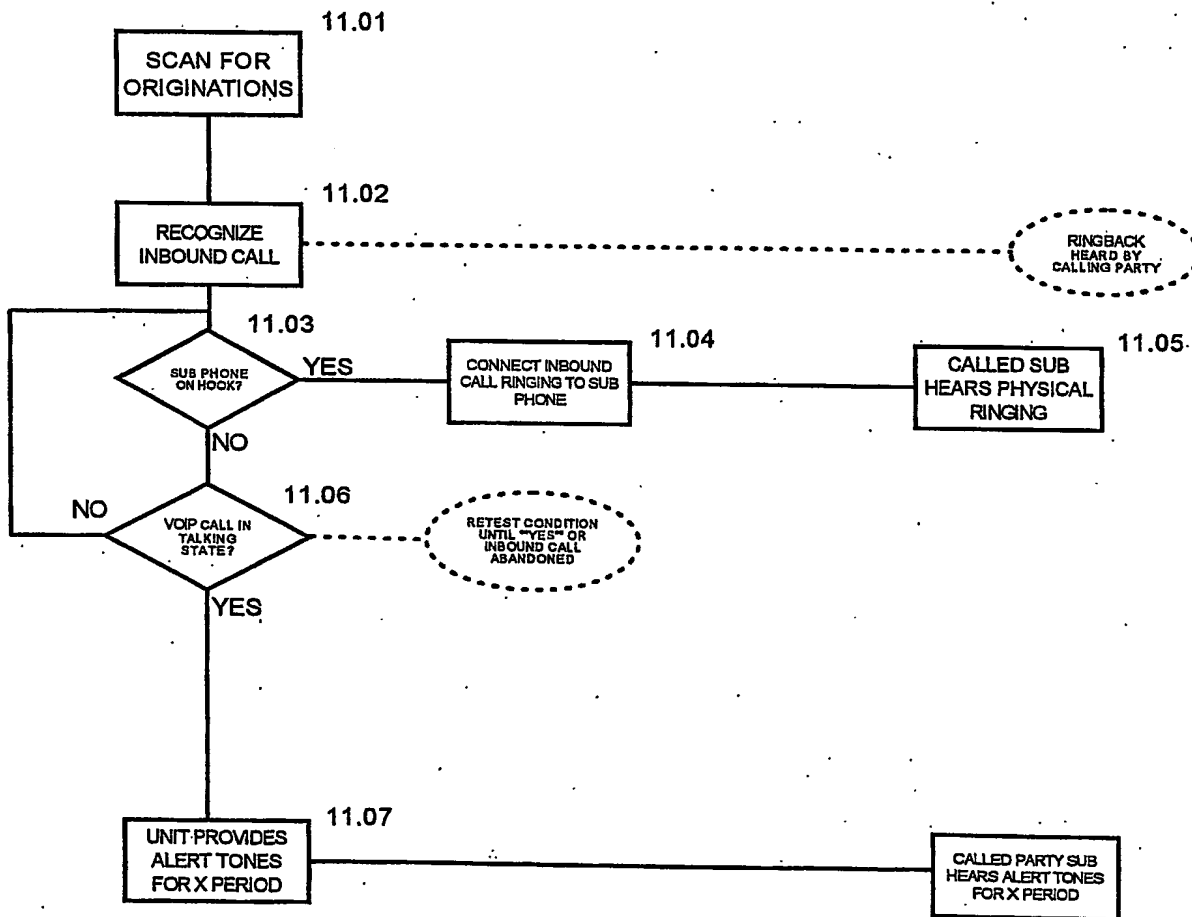


FIG. 11